FAST FOURIER TRANSFORMATION (FFT) WITH ADAPTION OF THE SAMPLING RATEIN DIGITAL RADIO MONDIALE (DRM) RECEIVERS

Background

5   Digital Radio Mondiale (DRM) is a new international standard for radio broadcasts at frequencies below 30MHz described in Digital Radio Mondiale (DRM): System Specification, ETSI TS 101 980 V1.1.1, September 2001. At the heart of this transmission system is the OFDM (Orthogonal Frequency Division Multiplex) modulated signal which is made up of a multitude of <u>uniformly spaced</u>

10   frequency carriers. Throughout the duration of each transmission symbol, the phase and amplitude of each carrier is fixed to signal a point in the QAM constellation. The position of the constellation point is chosen to represent a series of data bits. These data bits originate from the data payload (compressed digital audio, binary image data, etc.) after channel error control coding and interleaving have been applied. The complete encoding procedure and modulation method are already defined in the DRM standard.

In DRM, the number of carriers and the length of each symbol vary with the transmission/robustness modes and the broadcast bandwidth to suit a wide range of broadcast environments:

Robustness mode	Carrier number	Channel bandwidth (kHz)					
		4.5	5	9	10	18	20
A	min.	2	2	-102	-114	-98	-110
	max.	102	114	102	114	314	350
B	min.	1	1	-91	-103	-87	-99
	max.	91	103	91	103	279	311
C	min.				-69		-67
	max.				69		213
D	min.				-44		-43
	max.				44		135

Robustness mode	Duration Tu	Carrier spacing 1/Tu	Duration of guard interval Tg	Duration of symbol Ts = Tu + Tg	Tg/Tu	Number of symbols per frame
A	24 ms	41 2/3 Hz	2.66 ms	26.66 ms	1/9	15
B	21.33 ms	46 7/8 Hz	5.33 ms	26.66 ms	1/4	15
C	14.66 ms	68 2/11 Hz	5.33 ms	20 ms	4/11	20
D	9.33 ms	107 1/7 Hz	7.33 ms	16.66 ms	11/14	24

Table 1 DRM signal characteristics

5   (Carrier number 0 is at the centre frequency of the d.c. component in a conventional A.M. channel.)

Figure 1 shows the basic components of a typical DRM receiver including antenna 1 providing RF input to channel selector and RF downconverter 2. The resulting intermediate frequency is supplied to A/D converter 3 and the digitised output is

10   supplied to I/Q separator (mixer) 4. Separator 4 supplies baseband I/Q carriers to channel filter 5 and the filtered output is subject to timing adjustment or resampling at stage 6 to be described in more detail below. The output of stage 6 is subject to fast Fourier transform (FFT) at stage 7, the output of which is supplied to AFC and deinterleaver 9. Deinterleaver 9 supplies constellation point

15   and channel state information to multi level coding (MLC) decoder 10, shown in more detail in figure 2. The output of MLC decoder 10 may be supplied to other decoders indicated at 11 as well as audio decoder 12 supplying D/A converter 13 which provides the audio output.

As shown in figure 2, MLC decoder 10 comprises metric generator 14, depuncturer 15 and Viterbi decoder 16. Decision feedback metric adaptation is optionally provided by convolutional encoder 17 and puncturer 18.

In a typical DRM receiver, after the translation of the received RF signal from the DRM broadcast frequency slot to the receiver's own intermediate frequency, the baseband signal is resolved into its in-phase and quadrature (I and Q) components, i.e. complex time domain samples. Depending on the design of the DRM demodulator, any number of the intermediate frequency (IF) stages may be used. If the final IF is high, the number of down-conversion stages is reduced, but possibly at the expense of more stringent performance requirements in the IF filter circuit design. In the extreme, the RF signal is not down-converted at all, and sampled directly after it has been suitably filtered to remove all out-of-band signals. The baseband signal is further separated into individual carrier frequencies, each of which bears a QAM constellation point. The coordinates of the constellation point determine the encoded data payload. Fourier transform is known as the process which transforms a continuous time signal to the frequency domain. To carry out the Fourier transform of a signal sampled at a constant interval, the Discrete Fourier Transform (DFT) is used.

The mathematic expression of the DFT is given as,

$$H_n = \sum_{k=0}^{N-1} h_k e^{2\pi kn/N}$$

where h_k represents the complex time domain baseband samples, and N is the number of carriers.

The relationship between the continuous frequency domain components and the numeric output of the DFT is given by,

$$H(f_n) \approx TH_n$$

where each carrier frequency

$$f_n = \frac{n}{N} \bullet \frac{1}{T} \quad \text{and} \quad n = -\frac{N}{2}, ..., \frac{N}{2}$$

and T is the sampling period.

The frequency components are equally spaced at $1/NT$. It is clear that the frequency spacing is directly proportional to the sampling rate of the input baseband signal. Also note that in different Robustness Modes of DRM, the carrier spacing is different, and there is no simple harmonic relationship.

Using the formulation of the DFT shown above, the computational burden of a direct evaluation on the baseband signal (which is already windowed by the length of the transmission symbol N) via convolutions is non-trivial. There are known techniques for performing a DFT with reduced computational demands. The classical choice is the so-called Fast Fourier Transform (FFT) made famous by Cooley and Tukey, see "An Algorithm for the Machine Calculation of Complex Fourier Series", J.W. Cooley and J.W. Tukey. Math. Of Computation, issue 19, 1965. This paper documented the rediscovery of the FFT technique which was conceived in many early texts. This method is highly efficient in both hardware and software implementations due to its symmetrical and regular structure. However, the number of its input (time domain signal samples) and output (complex frequency components) are both restricted to integer exponentials of 2, i.e. N must be an integer exponential of 2.

It is worth mentioning that there are other efficient DFT techniques such as the Winograd Fourier Transform (On Computing the Discrete Fourier Transform, S. Winograd. Math. Of Comp., issue 32, 1978) (the number of frequency bins are restricted to a few fixed values, and complicated hardware implementation), and the Prime Factor FFT (A Prime Factor FFT Algorithm Using High-Speed Convolution, D.P. Kolba and T.W. Parks. IEEE Trans. ASSP, vol.25, no.4, August 1977) (requires breaking down of the long transform into shorter prime factored ones). They are not widely used because of their inherent restrictions.

Summary of the Invention

The present invention provides a method as described in annexed claim 1. Thus, the signal is modified at stage 6 before being supplied to the FFT 7. Preferably, the sample rate is an integral multiple of the desired frequency spacing in the

5   Fourier transform output. This can be achieved either by use of a suitable sample rate or by interpolation to provide more samples and thereby supply a larger apparent sample rate to the Fourier transform.

An embodiment of the invention will now be described by way of example only. In the drawing:

10   Figure 1 is a schematic diagram of the basic components of a DRM receiver; and

Figure 2 shows the components of a MLC decoder.

Detailed Description

Embodiments of this invention will now be described by way of example.

15   As can be seen in Table 1, the number of OFDM carriers is never an integer power of 2. On the surface this would suggest that it is not suitable to deploy the Cooley and Tukey FFT to evaluate the DFT efficiently.

Key point 1 An alternative approach is to use a bigger FFT which would generate more frequency components than the number of carriers, and then discarding the

20   unwanted ones. This would appear somewhat wasteful, but the computational benefits from using the FFT still outweigh direct DFT evaluation. To use the FFT, the number of input samples to the FFT is chosen to be equal to the next integer exponential of 2 above the number of frequency carriers. For example, if the number of complex carriers is 205, as in Robustness mode A at channel

25   bandwidth 9kHz, then the number of complex input samples (N) would be 2^8or

256. In order to keep the frequency spacing exactly 41 2/3 Hz, the sampling rate would need to be (256 x 41 2/3) = 10666 2/3 Hz.

In order to use the FFT, the number of complex time domain samples from the symbol needs to be exactly the same number of complex frequency bins. This

5   would   imply   a   number   of   different   sample   rates   for   different transmission/robustness modes and channel bandwidths.

As the carrier spacing is fixed within each Robustness mode, the highest sampling rate required in a particular mode is determined from the maximum number of carriers.   (To be more accurate, the maximum number of carrier should be

10   rounded up to the next integer exponential of 2.) The values for each Robustness mode are tabulated below. Note that the centre frequency of the transmission is always fixed at carrier number $n = 0$. Hence the maximum number of carriers to be generated by the FFT is twice the integer exponential of 2 necessary on the side of the transmission band with more carriers. For example, in Robustness mode A

15   at 20kHz channel bandwidth, the carrier numbers on the two sides are -110 and 350. On the positive side, the minimum integer exponential of 2 is 512, and thus the size of the FFT is twice that at 1024.

Robustness mode	Maximum number of carriers (rounded up)	Carrier spacing 1/Tu	Sampling rate
A	1024	41 2/3 Hz	42666 2/3 Hz
B	1024	46 7/8 Hz	48000 Hz
C	512	68 2/11 Hz	34909 1/11 Hz
D	512	107 1/7 Hz	54857 1/7 Hz

Table 2 Standard sampling rates

20   Given that N is always a power of 2, if fewer carriers are required to be evaluated (due to a narrower channel bandwidth), then the number of samples to include in each FFT can be reduced by decimating systematically at 1:2, 1:4, 1:8, etc.

Key Point 2 In practical implementations, only a single or a limited number of fixed sample rates would be available. However, a variety of techniques are available to workaround the sample rate restriction:

1. Interpolation of received signal in the time domain. This may be done by polyphase filtering or other means.

2. Over-sample the received signal at a rate which is the lowest common multiple (LCM) of all the different ones, and then decimate accordingly to accommodate different FFT sizes and frequency spacing. For example, the LCM for the standard sampling rates of the four Robustness modes as shown in Table 2 is 384000 Hz. To obtain the standard rate samples in Robustness mode A, a decimation factor at 1:9 is required, i.e. one sample is accepted in every 9. Similarly for Robustness modes B, C and D, the decimation factors are given as 1:8, 1:11 and 1:7 respectively.